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Abstract: The rising number of wireless users and terminals has pushed the deployment of many applications over wireless networks. Video multicasting is one of them. Wireless multicast has a great benefit in terms of resource utilization. Due to wireless nature of the media, a packet that is sent only once can reach all recipients. However, multicasting lacks in feed back mechanism. This makes it hard to deal with reliability or quality of service. Moreover, in order to reach all nodes especially the farther ones, multicast is always sent at the basic rate of 1 or 2 Mbps. This low rate may penalize other traffics and waste bandwidth capacity because of longer channel occupancy. As IEEE 802.11 standard provides possibility of multi-rate transmission, we propose to adapt multicast transmission rate according to quality of experience perceived at multicast users. We illustrate the significant performance improvement obtained with our scheme comparing to other existing schemes.

Key-words: Rate Adaptation Mechanism, Multicast, Video Streaming, Wireless Networks, Quality of Experience

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Mécanisms d'Adaptation de Débit pour Multimédia Multicast dans IEEE 802.11

Résumé : Le nombre croissant d'utilisateurs des réseaux et terminaux sans-fil a poussé le déploiement de nombreuses applications sur ces types de réseaux. Video multicasting est l'un d'eux. Multicast sans fil a un grand avantage en termes d'utilisation des ressources. En raison de la nature sans-fil du média, un paquet qui est envoyé une seule fois peut atteindre tous les bénéficiaires. Toutefois, la multicasting manque de mécanisme de feed-back. Il est donc difficile de faire face la fiabilité ou la qualité du service. En outre, afin d'atteindre tous les noeuds en particulier les plus loins, le multicast est toujours envoyé au taux de base de 1 ou 2 Mbps. Ce faible taux peut pénaliser d'autres trafics en gaspillant la capacité de bande passante cause de plus long temps d'occupation du canal. Puisque la norme IEEE 802.11 prévoit une possibilité de transmission multi-débit, nous proposons d'adapter les taux de transmission multicast en fonction de la qualité perue au multicast utilisateurs. Nous illustrons l'importante amélioration de la performance obtenue avec notre systme par rapport d'autres systmes existants.

Mots-clés : Mécanism d'adaptation de débit, Multicast, Video Streaming, Réseaux sans-fil, Quality d'experience

1 Introduction

Communication in the Internet can be done via different transmission modes. A classification has separated transmission into three modes: unicast, multicast, and broadcast. Unicast is one-to-one mode while broadcast is one-to-all mode. Multicast is in the middle and it can also be considered as a special type of broadcast because it is also one-to-many mode. More precisely, multicast is the transmission of packets to multiple destinations simultaneously.

Multicast over wireless networks is a fundamental communication function because wireless network is inherently broadcast-oriented. This means that a packet can be intercepted by all nodes in the sender's transmission range. Hence, each packet is sent just once and will reach all intended recipients in the multicast group. Therefore, multicast is an efficient method to send packet to a group since it allows transmission of packets to multiple destinations using fewer network resources. Moreover, the fast-growth of wireless network and its application has pushed the deployment of multicast communication over wireless networks. Various applications support multicast, for example, conference meeting, mobile commerce (mobile auctions), military command and control, distance education, entertainment service, and intelligent transport systems.

However, multicast application has some constraints. Multicast traffic has been set to the lowest transmission rate (basic rate) in order to reach all mobile nodes especially the further ones because they are subject to important signal fading and interference. The lower rates disadvantage transmission in terms of channel occupancy since they take longer time to send packets. This performance anomaly has been presented in [1]. Another constraint in multicast transmission is the lack of acknowledgment (ACK) and retransmission due to huge amount of traffic overhead these packets will generate. This is severe when transmission mode is multicast. First, the number of ACK will be multiplied by the number of recipients in the multicast group, which could cause collision due to ACK implosion. Second, it is difficult to deal with synchronization in the group if sender has to handle retransmission with per-connection basis.

Besides, multicast transmission is not reliable, losses can occur. In wireless networks, there are principally two types of losses due to two factors: *Bit Error Rate (BER)* resulted from signal strength and physical modulation, it can be called as *channel error*; and *Collision* resulted mainly from congestion in the network. It can be noticed that rate adaptation, a mechanism that switches transmission rate to improve performance, is not efficient in lossy network caused by the second factor as the network is already high-loaded and it would not perform better if we slower down the transmission. On the other hand, rate adaptation can be helpful in the first case when losses are caused by BER due to bad channel condition (path loss, interference, distance, etc). Therefore, this paper deals mainly with this type of loss, which is essential in wireless multicast communications.

As to improve performance of multicast transmission (loss rate, network utilization, and user perception), we apply rate adaptation mechanism using quality of experience as an indicator for transmission rate selection. Quality of Experience (QoE) is a new concept, more appropriate to multimedia service such as video or voice over IP. With these applications, quality of service is hardly determined only by technical parameters such as BER, SNR, etc... It makes more sense to evaluate quality by users' opinion on their perception of the application that is why it is called *Quality of Experience*. This metric can be evaluated in terms of Mean Opinion Score (MOS) shown in Table 1.

Table 1: Mean Opinion Score - MOS

MOS	**Quality**	**Impairment**
5	Excellent	Imperceptible
4	Good	Perceptible but not annoying
3	Fair	Slightly annoying
2	Poor	Annoying
1	Bad	Very annoying

However, it is difficult to ask people to evaluate the score and then adjust the transmission rate in real-time. The evaluation procedure is very complex and time-consuming, it also needs manpower. Thus it is not practical for real-time usage. For these reasons, in this paper we use Pseudo Subjective Quality Assessment (PSQA) [2], a real-time QoE assessment tool based on Random Neural Network, to evaluate QoE and we adapt transmission rate accordingly.

The rest of this paper is organized as follow. We begin by giving description of related works concerning rate adaptation mechanism in IEEE 802.11 wireless networks in Section II. We continue with the proposed scheme based on quality of experience in Section III. We explain implementations, scenario, and threshold selection in Section VI. The results are given in Section V where we also give comparisons of our schemes to others existing ones. Finally, we give conclusions and future works in Section VI.

2 Related Work

In this section, we begin by some related works concerning rate adaptation mechanisms in IEEE 802.11 namely Auto Rate Fallback (ARF), Receiver based Auto Rate (RBAR), and Adaptive ARF (AARF). Then we continue with rate adaptation mechanisms designed especially for multicast in wireless networks.

2.1 Rate adaptation mechanism in IEEE 802.11

In IEEE 802.11, the signal strength or SNR (Signal to Noise Ratio) obtained at receiver is linked to transmission rate of access point. The higher transmission rate implies the higher SNR. However, when distance between the access point and receiving stations increases, the reception condition degrades (due to interference, obstacles etc...), hence the need to switch to another rate. An order of magnitude for a mapping between distance and transmission rate in indoor scenario given by manufacturers can be approximated as in Table 2.

Table 2: ARF mapping between distance and bandwidth in 802.11b

Bandwidth (Mbps)	11	5.5	2	1
Distance (m)	0-50	50-75	75-100	100-150

- ARF-Auto Rate Fallback

  In ARF, when SNR decreases, an access point tries to recover by decreasing the bandwidth to 5.5, 2, and 1 Mbps respectively. The AP switches to a higher rate when a certain number (10) of packets has been successfully received; it switches back to the lower rate when a failure occurs. One problem of ARF is the usage of fixed threshold-based mechanism, which cannot adapt efficiently in varying wireless environment.

- RBAR- Receiver-Based Auto Rate

  RBAR [3] has the goal of performance optimization in wireless networks using also a rate adaptation protocol in MAC layer. In RBAR, RTS/CTS mechanism is enabled in order to get/send feedback from receiver. Before each transmission, RTS is sent out and is received by the receiver who computes and sends the transmission rate (in CTS) for the access point to use for the next transmission based on SNR. RTS and CTS headers have been modified for the purposes. This mechanism is based on SNR (computed with a priori channel model), which is a physical parameter that does not always correlate well with human perception. Moreover, RTS/CTS mechanism is not enabled in multicast.

- AARF-Adaptive ARF

  In AARF [4], the authors also use threshold-based mechanism but instead of setting it to a fixed number, threshold follows binary exponential backoff continuously at runtime to better reflect to the channel conditions. This means they multiply by two the number of consecutive successful transmission required to switch to a higher rate. The effect of this adaptation mechanism is to increase the period between successive failed attempts to use a higher rate. With fewer failed transmissions and retransmissions, the overall throughput is improved. Eventhough AARF is an efficient adaptation mechanism, unfortunately it cannot be applied to multicast scenario since the implementation of rate adaptation also includes the existence of acknowledgment and retransmission, which are disabled in multicast communication.

2.2 Rate adaptation in wireless multicast

Rate adaptation mechanism in multicast is different than in IEEE 802.11. The critical concern is the lack of feedback mechanism from receivers (no ACK or NACK) and also no retransmission to recover from loss/error. Many researchers have proposed reliable multicast protocols such as [5] or [6] to deal with unreliability issue in multicast. Another issue is performance of multicast due to the fact that multicast transmission rate is set to basic rate. Very few papers deal with this problem using multi-rate capability available in 802.11. We give brief descriptions of two of them below.

- RAM-Rate Adaptive Multicast

  In this protocol [7], multicast receivers make use of RTS to measure channel condition and send back transmission rate for sender to use in CTS. In case that a member does not receive the data frame correctly, it will send a NACK (Not Acknowledge). For enhancing the throughput, the authors added a frame sequence filed to RTS. This field is used by the member to check whether multicast data frame is a new frame or retransmission. If a frame is a retransmission of a previously successfully received frame, a member will not participate in this multicast transmission. This reduces the number of retransmission. It can be noticed that the protocol makes use of RTS/CTS and NACK, which is disable in multicast. Moreover, there are many modifications to existing frames.

- SARM-SNR-based Auto Rate for Multicast

  Park et al. have proposed SARM (SNR-based Auto Rate for Multicast) [8], a MAC-layer multicast mechanism with a multi-rate transmission. By changing multicast transmission rate on the basis of SNR values reported by mobile nodes, the wireless channel is used more efficiently than the default rate. In fact, SARM adapts a transmission rate according to the SNR of the node experiencing the worst channel condition. The SNR references are obtained from a table listing required SNR for PSNR to be higher than 30 (representing good QoS) for each transmission rate. Due to the lack of feedback mechanism in multicast, this scheme uses channel probing mechanism to inform the access point of the channel quality at mobile nodes. To avoid collision when nodes transmit feedback to the access point, the author also proposed a backoff timer for each mobile node based on the received SNR. Since this scheme has the closest goal to ours, we decide to compare it in our results.

3 QoE-based Rate Adaptation Mechanism

Previous algorithms always deal at packet level. In this paper, we propose a novel rate adaptation mechanism that handles the problem at the user-end perception level in terms of quality of experience. The idea of the proposed scheme is to use feedback from mobile stations to provision the current QoE of the network. For that, communication between access point and mobile nodes is needed. We make use of IEEE 802.11k standard [9] since the standard has specified many measurement requests and reports that can be used. It can be noticed that with 802.11k measurements, our control traffic is not significant in terms of overhead as we send control traffic much less frequently than packet-level schemes (e.g. one ACK for every single packet).

3.1 Pseudo-Subjective Quality Assessment

In order to get QoE in real-time, we deployed PSQA (Pseudo-Subjective Quality Assessment) [2], which is based on statistic learning using random neural network (RNN). The idea is to train the RNN to learn the mapping between QoE score and technical parameters so that we can use a trained-RNN as a function to give QoE score in real-time. In order to use this tool, three steps need to be done a priori. We summarize them as follow.

1. Configuration: we first choose configurations, which are sets of quality affecting parameters such as codec, bandwidth, loss, and delay along with their ranges of values that will be used for the RNN training. Then we take

several video sequences to be distorted with the configurations previously chosen.

2. Training: we ask for a panel of human observer to evaluate the distorted videos (Fig. 1) and then we store the configurations and corresponding MOS into two databases: training and validation databases. After that, we train the RNN to learn the mapping of configurations and scores as defined in the training database. Once the tool has been trained, we have a function $f()$ that can map any value of parameters into MOS.

3. Validation: trained RNN is validated by comparing value given by the $f()$ at the point corresponding to each configuration in the validation database. If the values are closed enough, the RNN is validated; otherwise, we have to review chosen configurations.

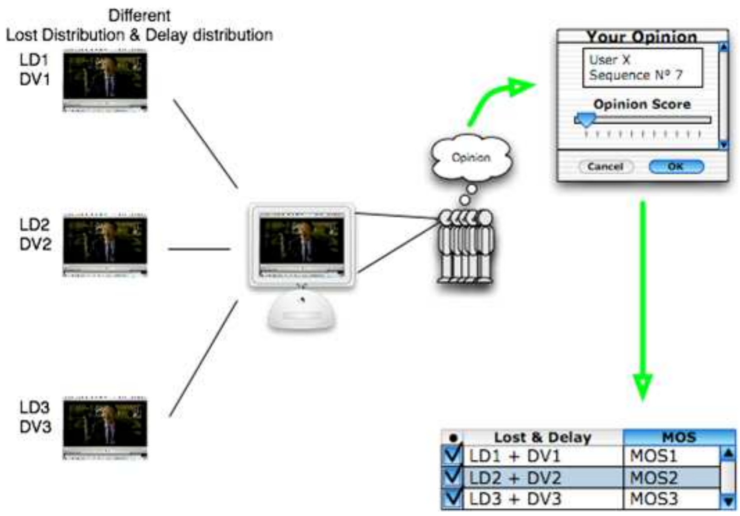

Figure 1: Subjective Quality Assessment

Once RNN has been validated, PSQA can be used anywhere in real-time without interaction from human. It needs values of technical parameters as input and it gives scores (in MOS) as if there were real humans marking the playing media.

3.2 Algorithm

We propose to use QoE as indicator for switching from one transmission rate to another. This is because we found out that for multimedia application such as video multicasting, it is more reasonable to adapt the transmission rate taking into account the quality perceived at the user end. We assume PSQA running on every multicast node.

We describe in Fig. 2 the behavior of an access point in our scheme during multicast session. At the beginning, the access point transmits multicast traffic at its highest rate. The AP monitors its attached clients every monitoring interval (mi). Note that our scheme uses time scale in terms of second because

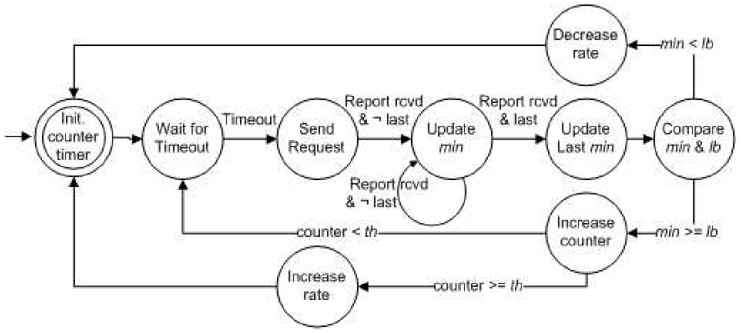

Figure 2: Access point behavior during multicast session

this scale is more reasonable than scaling in packet when dealing with human perception. When the timer rings, AP begins by sending requests to multicast members in order of membership precedence. This is to avoid collision of reports sending back from members. When a report is received, AP updates the minimum MOS (*min*) of the group accordingly. Once the last report has been received, it compares *min* with the lower bound (*lb*). This *lb* is computed by adding a margin (*mg*) to a reference score (*rf*), which is an acceptable score for the application. If *min* is less than *lb*, then AP switches immediately to one-step lower rate until minimum rate. If *min* is higher than *lb*, then AP increases the counter (representing the duration that AP has been waiting). If the counter reaches a threshold (*th*), then AP switches to one-step upper rate until maximum rate.

It can be noticed that when condition degrades, the AP in our scheme lowers transmission rate immediately. This is to adapt instantly to bad condition because it is essential to recover from the bad state rapidly. When network condition becomes better (i.e *min* is higher than *lb*) for several times, AP switches to higher rate. This waiting threshold is used to avoid ping-pong effect, before sending at higher rate (higher risk of BER), we make sure that this condition remains quite stable.

The important issue we have to consider is the choice of threshold value. It can be noticed that if *th* is too high, the system is less robust and may not respond quick enough to the current situation in the network. This may also result in the waste of bandwidth because before switching to higher transmission rate the AP has to wait very long time. However, this high value of *th* ensures that the condition in the network is stable before changing to the higher rate thus higher risk of BER. On the other hand, if *th* is too small, the system will be robust and react rapidly to the current condition. Therefore the AP can switch quicker to higher rate and profit from higher throughput. Nevertheless, if *th* is too small, the AP changes transmission rate often and this may result in ping-pong effect that make AP changing transmission rate all the time consuming more computations. Another consideration when deciding *th* is the fact that we should also consider the duration of the video and the session to be reasonable with the threshold. We make simulations in order to select appropriate threshold in Section 4.3.

4 Simulation Setup

In this section we begin by explaining the implementation of our scheme in NS2. We continue with description of our scenario. Then we gives results about threshold selection issue mentioned in algorithm section.

4.1 Implementation

We are interested in wireless networks IEEE 802.11 operates in infrastructure mode meaning that all traffic passes through an access point. The video sequence is an H.264-coded sequence of duration 60 seconds. It is encoded at 384 Kbps and streamed in multicast mode using UDP. Our implementation has been done via the network simulator NS-2 version 2.29 [10]. We patch wireless IEEE 802.11 implementation flaws of the original version with wireless update patch from [11]. The patch includes realistic channel propagation, Ricean propagation model, 802.11 bug fixes, multiple data transmission rates support, and adaptive auto rate fallback (AARF). We implement video streaming application by adding a video packet transmission module in NS-2. For communications between PSQA and NS2, we have integrated PSQA into NS2 so that it can get required statistics input for RNN. We adapt modulation in realtime according to PSQA score using our algorithm.

4.2 Scenario

Fig. 3 illustrates our topology, there is one video server on the Internet with multicast nodes connected to it via an access point. For our test, we use video with encoding rate illustrated in Fig.4. According to the video encoding rate and limited bandwidth in basic rate, we do not put many multicast nodes in order not to be biased concerning throughput issue. We decided to test with three multicast nodes. At the beginning, all nodes locate near by the AP (less than 50m radius). At 10s, station1 (st1) moves away from the access point (150m), and then at 40s it begins to move back to its initial position.

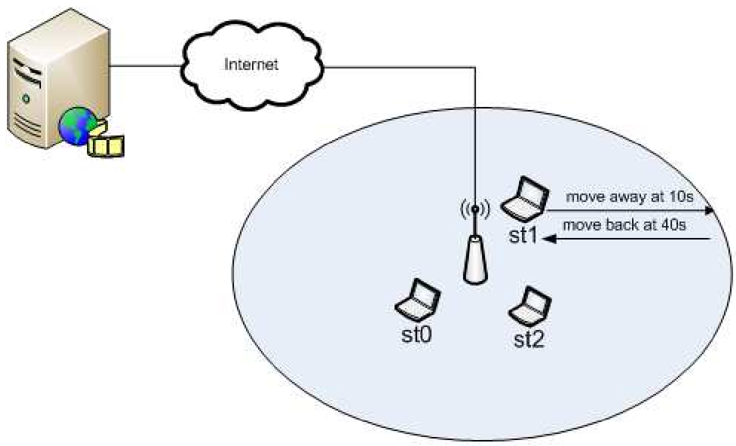

Figure 3: Topology of our scenario

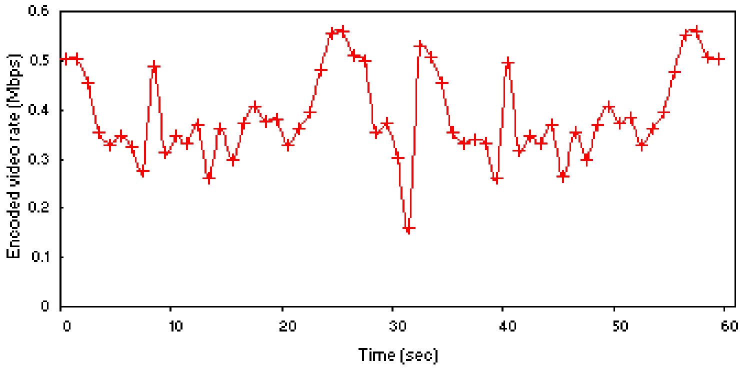

Figure 4: Rate variation of encoded video

4.3 Threshold Selection

For simulating different values of threshold (knowing that we set *mi* to 1 second), we select to test 8 different values ranging from 1 to 8. We illustrate in Fig. 5 and Fig. 6 the quality of experience and the goodput obtained with different values of threshold. Please note that the curves in Fig. 5 are normalized, this means that the results are divided by maximum value which is MOS=5. Both graphs are stacked meaning that the values have been shifted by x which is equal to i - 1 where i is the value of threshold. Since the curves have quite similar trends we also give summaries of average QoE and goodput values (of all connections) for each threshold in Table 3 and 4 respectively. We have found that surprisingly the goodput variation is not effected that much if we consider the whole connection duration. So we focused on the duration while one node is in movement (during 20s to 40s) and we can see the different as shown in Table 5 and 6.

With all the arguments seen from the experiments, we have selected value 5 for *th* because it is a compromised value that gives reasonable reactivity while maintaining high MOS and goodput. Therefore the simulations in the next part have been done with: mi=1 second, th=5 second; rf=3 (acceptable MOS for video application), mg=1, thus lb=4 as in Table 7 below.

Table 3: QoE obtained by different values of threshold for the whole connection

Threshold	1	2	3	4	**5**	6	7	8
MOS	3.89	4.39	4.41	4.48	**4.49**	4.51	4.59	4.62

Table 4: Goodput obtained by different thresholds for the whole connection

Threshold	1	2	3	4	**5**	6	7	8
Mbps	1.07	1.10	1.06	1.07	**1.08**	1.09	1.09	1.07

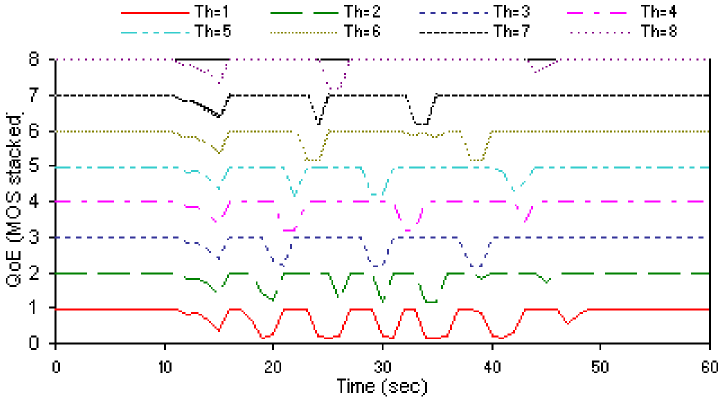

Figure 5: Quality of experience for different values of threshold

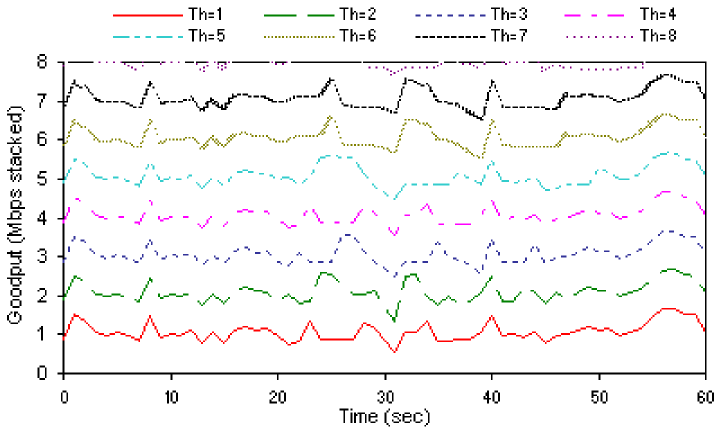

Figure 6: Goodput for different values of threshold

Table 5: QoE obtained by different values of threshold during mobility

Threshold	1	2	3	4	**5**	6	7	8
MOS	2.98	3.92	3.8	4.13	**4.32**	4.08	4.32	4.51

Table 6: Goodput obtained by different thresholds during mobility

Threshold	1	2	3	4	**5**	6	7	8
Mbps	0.98	1.05	0.97	0.98	**1.05**	1.05	1.05	1.06

Table 7: Configuration of our parameters

Parameter	Description	Value
mi	monitoring interval	1
th	threshold	5
rt	reference score	3
mg	margin	1
lb	lower bound	4

5 Results

We illustrate the results with two metrics: the goodput (for network utilization) and QoE (for user perception). We compare our scheme to three different approaches: default multicast (with 1 Mbps), maximum rate (with 11 Mbps), and SARM-like (SNR-based).

5.1 Goodput

We first illustrate in Fig.7 the average goodput obtained from each scheme. Then we detail to see how an individual station behaves in terms of goodput. For that we show two more graphs concerning a fixed station (st0) located near by the AP in Fig. 8 and a moving station (moving away from and back to the AP) in Fig. 9.

Observation from Fig.7:

- When the node moves (during 10s to 40s), our goodput is much higher than all others because it adapts to user perception (resulted from all parameters) directly.
- When transmit at default rate (1 Mbps), throughput is the lowest in general (graph before 10s and after 40s). This proves the problem of bandwidth wasting in multicast.
- Maximum rate's goodput is good but when the distance becomes farther, channel condition degrades. As we can see the degradation is obvious comparing to other schemes that have probably switched to lower rate.
- SNR-based performs better than default multicast rate, which is conformed to what have been done in [8]. However, SNR in our scenario is quite low because of mobility and this makes the scheme changing to lowest rate as we can observe in the graph; when the mobile station begins to move, the scheme behaves the same way as in default-1M.

We can see from Fig.8 that for a fixed station located nearby the AP, its goodput does not change that much. Its variation is due more to the encoding rate (see with Fig.4) than the channel condition. However, we can observe that using 11M for transmission gives a little higher advantage in terms of goodput. This is because when the station is closed to the AP, it can profit efficiently from short distance and high transmission rate. On the contrary, for a moving station in Fig.9 its goodput varies often during station's movement. We observe

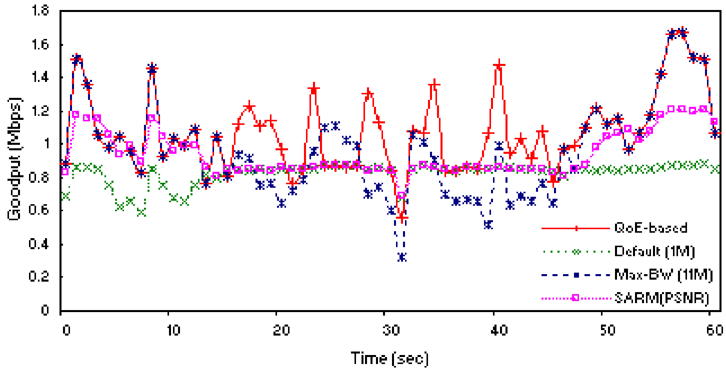

Figure 7: Average goodput of all stations

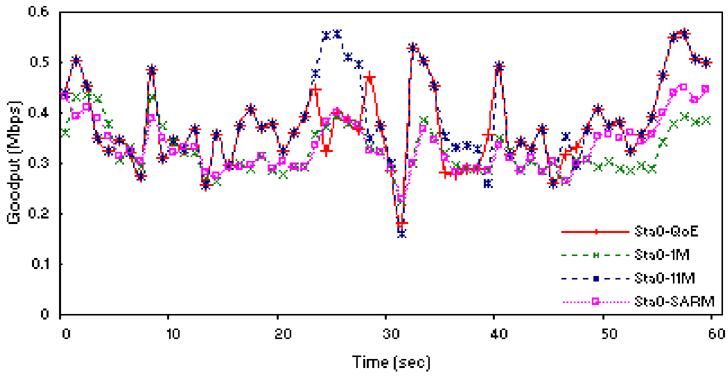

Figure 8: Goodput of a fixed station

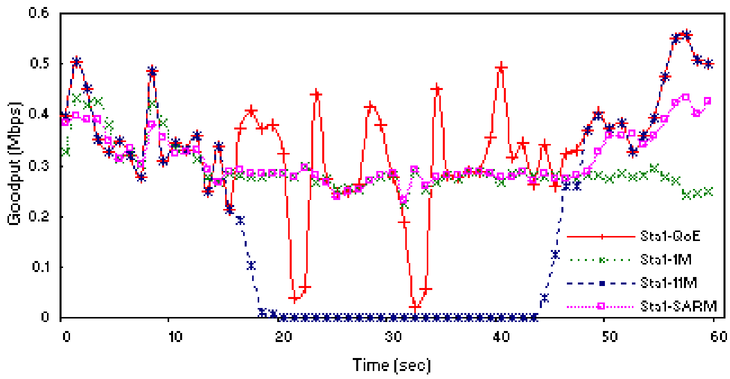

Figure 9: Goodput of a moving station

few drops in our scheme due to the time used to switch to lower rate. We also observe that using high transmission rate (11M) giving very bad results this is due to the high BER the station suffered when moving away from the AP.

Note that we illustrate here only the goodput of multicast traffic. It can be seen that if we consider also background traffic, its goodput will be increased when the rate increases (see rates variation in Fig.10) and we gain more goodput as much as we stay at higher rates.

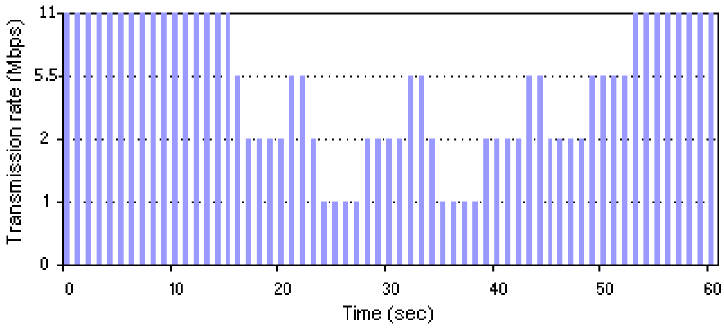

Figure 10: Rate adaptation during the test scenario

5.2 Quality of Experience

We illustrate two graphs concerning minimum QoE in time and average QoE of all stations.

Fig. 11 illustrates the scores obtained by a member encountered the worst channel condition. During moving period, we can see that all schemes experience quite bad performance. The worst scheme is maximum-11M because the rate is too high, and then follows by SNR-based and Default-1M respectively. Despite that our scheme performs the best, we also have some drops caused by the time taken to adapt to the bad channel condition. Fig. 12 illustrates the overall performance of the network. Since we use QoE as indicator in our scheme, we get a great performance in terms of QoE (the average QoE is at least 3.5). However, there are a few drops in the graph due to the time our scheme uses to adapt to the new condition. We also observed that the main problem of SARM-like mechanism may be caused by PSNR definition that does not have a direct relationship with QoE.

6 Conclusions and Future Works

We have proposed a rate adaptation mechanism based on QoE at receiver as it is the most important metric for multimedia application. We decided to use our mechanism to treat multicast performance problem because all the loss in multicast mainly resulted from channel error. Thus, adapting rate will improve performance. We have shown that our scheme has improved performances in terms of network utilization and user satisfaction in wireless multicast. In the future, we plan to investigate more on waiting threshold using strategies such as binary exponential backoff.

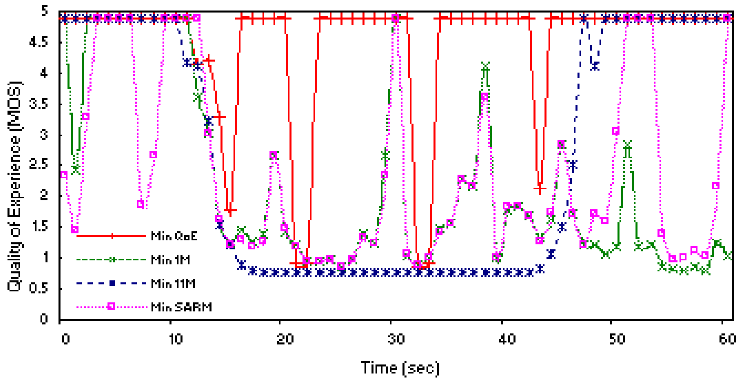

Figure 11: Minimum QoE during multicast for each scheme

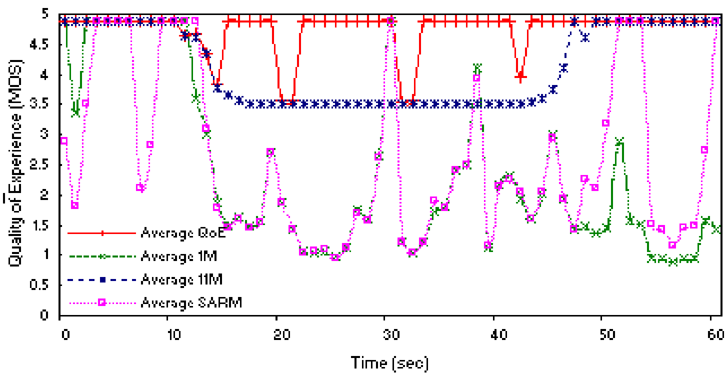

Figure 12: Average QoE of all stations for each scheme

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